

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

INVENTORS: Marcia R. Brumitt  
James M. Turnbull

ASSIGNEE: Jabra Corporation

SERIAL NUMBER: n/a

DATE FILED: n/a

TITLE: NOISE REDUCTION METHOD AND APPARATUS

ATTORNEY DOCKET: 4246 P

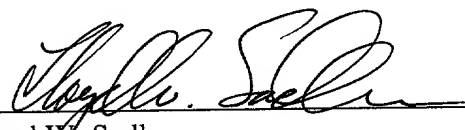
Assistant Commissioner for Patents  
Washington, DC 20231

**COVER LETTER**

Honorable Assistant Commissioner:

Enclosed herewith please find the following documents comprising a United States patent application: (1) specification, claims and drawings, (2) declaration of inventor(s), (3) fee calculation sheet, (4) fee, (5) statements of small entity status, (6) information disclosure statement and form(s) 1449, and (7) assignment and recordation cover sheet. Applicant intends to submit the copies of the references on the an information disclosure statement and form(s) 1449 as soon as copies of the relevant patents are available.

Respectfully submitted this 19<sup>th</sup> day of June, 2000.



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**VERIFIED STATEMENT (DECLARATION)  
CLAIMING SMALL ENTITY STATUS**

**--INDEPENDENT INVENTOR--  
(37 CFR 1.9(c), (f) and 1.27(b))**

Honorable Assistant Commissioner:

As the below named inventor, I hereby declare that I qualify as an independent inventor as defined in 37 CFR § 1.9(c) for the purposes of paying reduced fees under section 41(a) and (b) of Title 35, United States Code, to the Patent and Trademark Office with regard to the invention entitled **NOISE REDUCTION METHOD AND APPARATUS** described in a patent application filed herewith.

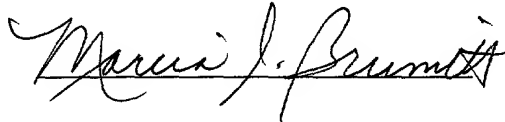
I have not assigned, granted, conveyed or licensed and I am not under any obligation under contract or law to assign, grant, convey or license any rights in the invention to any person who could not be classified as an independent inventor under 37 CFR § 1.9(c) if that person had made the invention, or to any concern which would not qualify as a small business concern under 37 CFR § 1.9(d) or a nonprofit organization under 37 CFR § 1.9(e).

I acknowledge the duty to file, in this application or patent, notification of any change in status resulting in loss of entitlement to small entity status prior to paying, or at the time of paying, the earliest of the issue fee or any maintenance fee due after the due date on which status as a small entity is no longer appropriate. (37 CFR § 1.28(b)).

I hereby declare that all statements made herein are of my own knowledge and are true and that all statements made on information and belief are believed to be true; and further that

these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application, any patent issuing thereon, or any patent to which this verified statement is directed.

Signature of Inventor:



Name of Inventor:

Marcia R. Brumitt

Date:

6/7/00

Signature of Inventor:



Name of Inventor:

James M. Turnbull

Date:

6/7/00

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Washington, DC 20231

**VERIFIED STATEMENT (DECLARATION)  
CLAIMING SMALL ENTITY STATUS**

**--SMALL BUSINESS CONCERN--  
(37 CFR 1.9(f) AND 1.27(c))**

Honorable Assistant Commissioner:

I hereby declare that I am

- ☐ the owner of the small business concern identified below:
- ☒ an official of the small business concern identified below and that I am empowered to act on behalf of said corporation:

NAME OF CONCERN: Jabra Corporation

ADDRESS OF CONCERN: 9171 Towne Centre Drive

San Diego, CA 92122

I hereby declare that the above identified small business concern qualifies as a small business concern as defined in 13 CFR § 121.3-18, and reproduced in 37 CFR § 1.9(d) for purposes of paying reduced fees under section 41(a) and (b) of Title 35, United States Code, in that the number of employees of the business concern, including those of its affiliates, does not exceed 500 persons. For the purposes of this statement, (1) the number of employees of the business concern is the average over the previous fiscal year of the concern of the persons

employed on a full-time, part-time or temporary basis during each of the pay periods of the fiscal year, and (2) concerns are affiliates of each other when either, directly or indirectly, one concern controls or has the power to control the other, or a third party controls or has the power to control both.

I hereby declare that rights under contract or law have been conveyed to and remain with the small business concern identified above with regard to the invention, entitled **NOISE REDUCTION METHOD AND APPARATUS** by the above-named inventors described in

- ☒ the specification filed with this declaration.
- ☐ application Serial No. \_\_\_\_\_, filed \_\_\_\_\_.
- ☐ Patent No. \_\_\_\_\_, issued \_\_\_\_\_.

If the rights held by the above-identified small business concern are not exclusive, each individual, concern or organization having rights to the invention is listed below and no rights to the invention are held by any person, other than the inventor, who could not qualify as an independent inventor under 37 CFR § 1.9(c) if that person made the invention, or by any concern which would not qualify as a small business concern under 37 CFR § 1.9(d), or a nonprofit organization under 37 CFR § 1.9(e).

- ☒ no such person, concern or organization exists.

each such person, concern or organization is listed below:

NAME: \_\_\_\_\_

ADDRESS: \_\_\_\_\_

INDIVIDUAL                      SMALL BUSINESS ENTITY  
NONPROFIT ORGANIZATION

I acknowledge the duty of the small business concern to file, in this application or patent, notification of any change in status resulting in loss of entitlement to small entity status prior to paying, or at the time of paying, the earliest of the issue fee or any maintenance fee due after the due date on which status as a small entity is no longer appropriate. (37 CFR § 1.28(b)).

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application, any patent issuing thereon, or any patent to which this verified statement is directed.

On Behalf of:

Jabra Corporation

NAME OF PERSON SIGNING:

Marcia Brumitt

TITLE OF PERSON SIGNING:

Chief Technology Officer

SIGNATURE:

Marcia Brumitt DATE: 6/7/00

- 1
- 2
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Be it known that Marcia R. Brumitt, a citizen of the United States of America, and M. Turnbull, a citizen of the United Kingdom, have invented a new and useful invention entitled NOISE REDUCTION METHOD AND APPARATUS of which the following comprises a complete specification.

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

10/25/99 U.S. PTO  
09/596700  
06/19/00

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Assistant Commissioner for Patents  
Washington, DC 20231

**CERTIFICATE OF MAILING**

Honorable Assistant Commissioner:

I hereby certify that the attached paper of fee (Patent Application) is being deposited with the United States Postal Service as "Express Mail Post Office to Addressee," Express Mail Mailing Label Number EK69229718105, (under 37 C.F.R. § 1.10) on this 19<sup>th</sup> day of June, 2000 with sufficient postage and in an envelope addressed to the Assistant Commissioner for Patents, Washington, DC 20231.



Signature of registered practitioner or other person  
having a reasonable basis to expect mailing to occur  
on the date of deposit shown pursuant to 37 C.F.R.  
§ 1.8(a) or person actually making deposit pursuant  
to 37 C.F.R. § 1.10(b).

Lloyd W. Sadler, Reg. No. 40,154

Name of person whose signature is given above



# NOISE REDUCTION METHOD AND APPARATUS

## Background of the Invention

Field of the Invention. This invention relates methods and apparatus' for reducing unwanted noise in a signal. More specifically, this invention relates to methods and apparatus' for reducing noise in a telephone speech communication signal.

Description of Related Art. A variety of different methods of signal noise reduction are well known in the art, however typically these previously methods introduce unwanted amplitude modulation or other audible artifacts to the resulting processed signal.

The reader is referred to the following U.S. and international patent documents for general background material: WO 89/06877, WO 95/25382, U.S. Patent No's: 4,061,875, 4,630,302, 4,811,404, 4,985,925, 5,036,540, 5,402,496, 5,490,233, 5,640,490, 5,848,171 and 5,970,441. Each of these patent documents is hereby incorporated by reference in its entirety for the material contained therein.

## Summary of the Invention

It is desirable to provide a method and apparatus for reducing the noise in a telephone or telephone-like communication system. For example, it is desirable to provide a method and apparatus that reduces the noise, either systematic or background, received when a computer operator/user employs voice recognition software and equipment to give voice commands to a computer system. The noise in this system can be induced by room noise such as other users, equipment and the like, or can be induced by communication equipment, fans, cross-talk, radio reception and the like. In this example, it is desirable to provide a method that may be performed within the computer

1 system. In an alternative example, it is desirable reduce the noise encountered by a  
2 cellular or PCS telephone system user in an automobile or other noisy environment. The  
3 noise in this example is caused by such sources as road noise, engine noise, and/or other  
4 acoustic sources such as the car radio. In this example, it is desirable to perform the  
5 noise reduction in the automobile telephone kit and will remove as much noise as  
6 possible before transferring the signal to the telephone for transmission. It is desirable to  
7 provide an apparatus and method for reducing noise in a telephone and/or telephone-like  
8 communication system.

9 Therefore, it is an object of this invention to provide a method and apparatus for  
10 reducing unwanted noise in a signal containing an information component and a noise  
11 component.

12 It is a further object of this invention to provide a method and apparatus for  
13 reducing unwanted noise in a signal that applies a time domain high frequency emphasis  
14 function.

15 It is another object of this invention to provide a method and apparatus for  
16 reducing unwanted noise in a signal that buffers an emphasized signal.

17 It is a still further object of this invention to provide a method and apparatus for  
18 reducing unwanted noise in a signal that applies a time domain windowing function to the  
19 buffered signal.

20 Another object of this invention is to provide a method and apparatus for reducing  
21 unwanted noise in a signal that converts windowed data from the time domain to the  
22 frequency domain to give frequency data in a number of frequency bins.

1           A further object of this invention is to provide a method and apparatus for  
2   reducing unwanted noise in a signal, with a spectral power calculated for each frequency  
3   bin.

4           A still further object of this invention is to provide a method and apparatus for  
5   reducing unwanted noise in a signal, where the overall or mean bin power can be  
6   optionally calculated.

7           Another object of this invention is to provide a method and apparatus for reducing  
8   unwanted noise in a signal, where the overall or mean bin power can optionally be  
9   limited to a minimal value.

10          Another object of this invention is to provide a method and apparatus for reducing  
11   unwanted noise in a signal, that temporally smoothes the spectral power results.

12          A still further object of this invention is to provide a method and apparatus for  
13   reducing unwanted noise in a signal, that transversally smoothes the temporally smoothed  
14   spectral power bins.

15          A further object of this invention is to provide a method and apparatus for  
16   reducing unwanted noise in a signal, that includes generating a weighting scalar for each  
17   bin based on two dimensionally smoothed spectral power bins and the optional overall or  
18   mean bin power, which may be limited.

19          It is another object of this invention to provide a method and apparatus for  
20   reducing unwanted noise in a signal, that includes multiplying the raw frequency bins by  
21   the weighting scalar.

1           It is a still further object of this invention to provide a method and apparatus for  
2   reducing unwanted noise in a signal that provides a conversion of the weighted frequency  
3   data from the frequency domain back into the time domain.

4           It is another object of this invention to provide a method and apparatus for  
5   reducing unwanted noise in a signal that uses a partial inverse window function.

6           Another object of this invention is to provide a method and apparatus for reducing  
7   unwanted noise in a signal, that applies a time domain high frequency de-emphasis  
8   function to provide a signal with reduced noise component, while maintaining an  
9   essentially unchanged information component.

10          A further object of this invention is to provide a method and apparatus for  
11   reducing unwanted noise in a signal, wherein the apparatus has an input for receiving an  
12   analog signal containing an information component and a noise component.

13          A still further object of this invention is to provide a method and apparatus for  
14   reducing unwanted noise in a signal, wherein the apparatus has a converter for converting  
15   an analog signal to a digital form.

16          Another object of this invention is to provide a method and apparatus for reducing  
17   unwanted noise in a signal, wherein the apparatus has a digital signal processor for  
18   performing such functions as pre-emphasis, buffering, windowing, Fast Fourier  
19   Transform, power calculations, temporal smoothing, transversal smoothing, generating  
20   weighting scalars, performing weighting of the frequency domain signal, Inverse Fast  
21   Fourier Transform, partial inverse windowing, and de-emphasis.

22          It is a further object of this invention to provide a method and apparatus for  
23   reducing unwanted noise in a signal, wherein the apparatus has non-volatile memory

1 containing program instructions for the digital signal processor to perform steps of the  
2 noise reduction method.

3 It is another object of this invention to provide a method and apparatus for  
4 reducing unwanted noise in a signal, wherein the apparatus has an output that converts  
5 the processed digital signal back into an analog form and which transmits the signal with  
6 the reduced noise component and essentially unchanged information component.

7 A further object of this invention is to provide a method and apparatus for  
8 reducing unwanted noise in a signal, wherein the apparatus has support circuitry as  
9 necessary for the digital signal processor and converters, including but not necessarily  
10 limited to a clock generator and a power supply.

11 A still further object of this invention is to provide a method and apparatus for  
12 reducing unwanted noise in a signal, where the apparatus may have on-board random  
13 access memory for storing digital signals, buffers and intermediate calculations.

14 These and other objects of the invention are achieved by the method and  
15 apparatus herein described and are readily apparent to those of ordinary skill in the art  
16 upon review of the following drawings, detailed description and claims.

### 17 **Brief Description of the Drawings**

18 In order to show the manner that the above recited and other advantages and  
19 objects of the invention are obtained, a more particular description of the preferred  
20 embodiments of this invention, which is illustrated in the appended drawing, is described  
21 as follows. The reader should understand that the drawings depict only present preferred  
22 and best mode embodiments of the invention, and are not to be considered as limiting in  
23 scope. A brief description of the drawings is as follows.

Figure 1 is a process flow chart showing the preferred processing steps of the noise reduction method of this invention.

Figures 2a and 2b are frequency plots demonstrating the frequency leveling effects of pre-emphasis.

Figures 3a and 3b are time domain plots showing the effect of pre-emphasis on the time domain waveform.

Figure 4 is a top-level simplified block diagram of buffer handling.

Figures 5a and 5b are plots of the Hanning and Inverse Hanning Window function.

Figure 6 is a plot of the typical and preferred weighting function of this invention.

Figures 7a and 7b are process diagrams showing snapshots of a speech sample without the smoothing functions applied.

Figures 8a and 8b are process diagrams showing snapshots of a speech sample with the smoothing functions applied.

Figures 9a-e are spectrograms of a speech sample showing the results of the process of this invention with various processing.

Figure 10 is a block diagram of the preferred apparatus of this invention for the cellular telephone embodiment.

Reference will now be made in detail to the present preferred embodiment of the invention, examples of which are illustrated in the accompanying drawings.

### **Detailed Description of the Invention**

Figure 1 is a process flow chart showing the preferred processing steps of the noise reduction method of this invention as well as the data flow between the processing steps. Initially, the noise cancellation algorithm receives 101 a digital data stream. The

1 digital data stream contains the signal that is to be conditioned by this invention. In its  
2 present preferred embodiment, this digital data stream can originate from an analog-to-  
3 digital converter, from a cellular telephone providing a digital voice output or the like.  
4 The resulting digital audio signal is passed through a pre-emphasis function 102, which  
5 flattens the spectral energy of the desired signal content. Typically, this desired signal  
6 content is a voice or speech signal, although alternative signal content can be used in this  
7 invention. By way of example, the spectral energy of a speech signal rolls off at  
8 approximately 6 dB per octave. This roll off can be compensated for by applying a  
9 difference function to the signal, since low frequency components of the speech signal  
10 typically have more signal energy than high frequency components.

11 If  $s(n)$  is the current speech sample and  $s(n-1)$  is the previous speech sample, then  
12 the frequency compensated signal  $s'$  is given by:  $s'(n) = s(n) - s(n-1)$ . Hence, the high  
13 frequency components of the signal are emphasized while the low frequency components  
14 are de-emphasized.

15 After the signal is pre-emphasized 102, consecutive, time domain, samples from  
16 the pre-emphasized input stream are stored 103 in a buffer for block processing. Next, a  
17 windowing function 104 is applied to the time domain data stored in the concatenated  
18 analysis buffer. The purpose of windowing 104 the time domain data prior to processing  
19 using a discrete Fourier transform method (such as a Fast Fourier Transform, or FFT) is  
20 to minimize spectral leakage. Spectral leakage occurs when a frequency component of  
21 the signal does not fall exactly centrally within a frequency bin. Energy from this  
22 component can spill into neighboring bins and beyond. The simplest windowing  
23 function, which has the greatest susceptibility to spectral leakage, is the Rectangular

1 window. A preferred and frequently used windowing function, which greatly reduces  
2 spectral leakage, is the Hanning window. A Fast Fourier Transform (FFT) step 105 is  
3 performed on the windowed 104 time domain data to transform the data into the  
4 frequency domain. The preferred FFT 105 size is  $2N$ . The resulting frequency domain  
5 buffer has  $2N$  frequency bins, each of which is a complex value.

6 Let  $F[0]$  represent the first bin and  $F[2N-1]$  represent the last bin. For further  
7 analysis, we are interested only in bins  $F[0]$  through  $F[N]$ , a total of  $N+1$  bins, which  
8 represents the positive frequency spectrum of the analyzed signal. Bins  $F[N+1]$  to  $F[2N-$   
9  $1]$  are further processed at a later stage of the method of this invention.  $F[n]$  is a complex  
10 number that comprises a real component  $Fr[n]$  and an imaginary component  $Fi[n]$ . The  
11 raw complex frequency data generated in the FFT 105 is passed to the Power Calculation  
12 block 106. The Power Calculation block 106 calculates an array of power estimates  $P[0 .$   
13  $. N]$  corresponding to each of the bins  $F[0]$  to  $F[N]$ , as follows:

$$14 \quad P[n] = Fr[n] * Fr[n] + Fi[n] * Fi[n].$$

15 If signal normalization is required later in the Weighting block 110, the overall  
16 frame power can be calculated as:

$$17 \quad P_t = P[0] + P[1] + \dots + P[N-1] + P[N].$$

18 The mean power per bin is calculated as:

$$19 \quad P_m = P_t / (N+1).$$

20 It is often desirable to apply normalization only to signals above a certain level, in  
21 which case the mean power,  $P_m$ , can be limited to a minimum value,  $P_o$ . If  $P_m$  is less  
22 than  $P_o$ , then  $P_m$  is sent to  $P_o$ . Signal normalization is usually necessary when the  
23 background noise and speech level change with time, such as is commonly found in an



1 automobile environment. When a car speeds up the background noise and, in particular,  
2 the road noise increases. When the level of background noise increases, the speaker  
3 automatically and naturally compensates by raising his or her voice. Fixed weighting  
4 thresholds do not tend to work particularly well in this situation. Where the background  
5 noise is somewhat constant, such as in an office environment, the speakers voice level  
6 does not tend to change substantially and, therefore, normalization may not be necessary  
7 in such an environment.

8 As further illustrated later in this specification, the power management of each bin  
9 can fluctuate dramatically from analysis frame to analysis frame. Note that when a plot  
10 of the power function for a particular bin is plotted against time it does not transition  
11 smoothly from one level to another. Rather, it fluctuates rapidly with time although it  
12 exhibits a general trend, which is seen to change more slowly with time. It is this  
13 relatively slow changing trend that is of particular interest in this invention. This high  
14 frequency like signal is superimposed on a low frequency signal, where the low  
15 frequency signal is the signal of interest. For this reason, a power array  $P[0 \dots N]$  from  
16 the Power Calculator 106 is applied to a Temporal Smoothing function 107, in which the  
17 data is smoothed with respect to time. Although simple averaging can be used, the  
18 preferred smoothing technique is to apply a first order digital low pass filter to each  
19 power bin. Therefore, in this invention a  $N+1$  low pass filters, each of which smoothes  
20 the power bins with respect to frame-to-frame fluctuations, is employed. The preferred  
21 first order low pass filter used for performing the temporal smoothing is of the form:

$$P_t[n] = A * P_{t'}[n] + B * P[n],$$

1 where  $P_t[n]$  is the temporally smoothed power for bin  $n$ ,  $P[n]$  is the raw power for bin  $n$ ,  
2 and  $P_t'[n]$  is the temporally smoothed power for bin  $n$  from the previous frame. For  $N$   
3 equal to 64, giving 128 point FFT analysis, and sampling at 8 kHz, it has been found  
4 through experimentation and observation that the preferred values for  $A$  and  $B$  are 0.75  
5 and 0.25 respectively give particularly good results.

6 As also illustrated in later in this specification, the power measurement for each  
7 bin can also fluctuate greatly from bin to bin; i.e., the power function plotted against bin  
8 number does not transition smoothly, rather it fluctuates rapidly as the bins are traversed  
9 with increasing frequency. However, the power function also exhibits a general trend,  
10 which is seen to change more slowly with bin number, and again it is this relatively  
11 slowly changing trend that is of interest in this invention. For this reason, the temporally  
12 smoothed data from the Temporal Smoothing block 107 is passed to a Transversal  
13 Smoothing Block 108. That is, once the successive frame results are visualized on a  
14 time-frequency plot, such as a spectrogram, the transversal smoothing is oriented  
15 transversally with respect to the temporal smoothing. Although a low pass filter could be  
16 used to perform the transversal smoothing 108, the preferred transversal smoothing  
17 technique 108 in this invention is to apply a simple averaging scheme. The preferred  
18 averaging function, which performs the transversal smoothing 108 is of the form:

$$19 \quad P_f[n] = (P_t[n-I] + P_t[n-I+1] + \dots + P_t[n] + \dots + P_t[n+I-1] + P_t[n+I]) / (2I + 1);$$

20 where  $P_f[n]$  is the transversally smoothed power for bin  $n$ ,  $P_t[n]$  is the temporally  
21 smoothed power for bin  $n$ , and  $I$  is the number of bins prior to and after the current bin of  
22 interest that the summation for the averaging will cover. For  $N$  equal to 64, giving 128  
23 point FFT analysis, and sampling at 8 kHz, it has been found through experimentation

1 and observation that a value of  $I$  of 3 gives particularly good results, and is therefore the  
2 preferred value.

3 The smoothed power data,  $Pf[0 \dots N]$  is passed to the Weighting Function  
4 Generator 109, which generates an array of weighting scalars  $W[0 \dots N]$ ,  $W[n]$  being a  
5 function of  $Pf[n]$  in the non-normalized case, or  $W[n]$  being a function of  $(Pf[n] - P_m)$  in  
6 the normalized case. The Weighting Function Generator 109 uses an array of scalars that  
7 will be applied to each frequency bin of the raw FFT data. The purpose of the weighting  
8 function is to leave the frequency bins with relatively large power levels unchanged and  
9 to attenuate the frequency bins with relatively low power levels. The reader is referred to  
10 figure 6 for a typical weighting function. The actual weighting is performed 110  
11 following the Weighting Function Generator 109, using data from both the Weighting  
12 Function Generator 109 and the FFT 105. Raw frequency values  $Fr[0]$  and  $Fi[0]$ , the real  
13 and imaginary components of  $F[0]$ , are multiplied by  $W[0]$ . Raw frequency values  $Fr[1]$   
14 and  $Fi[1]$  are multiplied by  $W[1]$ , and so on up to raw frequency values  $Fr[N]$  and  $Fi[N]$ ,  
15 which are multiplied by  $W[N]$ . To preserve the natural symmetry of the raw frequency  
16 data,  $Fr[N+1]$  and  $Fi[N+1]$  are multiplied by  $W[N-1]$ ,  $Fr[N+2]$  and  $Fi[N+2]$  are  
17 multiplied by  $W[N-2]$ , and so on up to  $Fr[2N-1]$  and  $Fi[2N-1]$ , which are multiplied by  
18  $W[1]$ . The weighted FFT data, of size  $2N$  complex values, is passed to the IFFT Block  
19 111, to give a time domain waveform of length  $2N$  real samples. The resulting waveform  
20 exhibits the same windowing applied by the Windowing block 104 and is passed through  
21 an Inverse Windowing block 112. The detailed characteristics of the preferred Inverse  
22 Windowing 112, is further described in relation to figure 5. This Inverse Windowing  
23 block 112, de-windows the center  $N$  samples of the frame to give a time domain sample

1 of length  $N$ , which does not have any amplitude modulation. In the preferred  
2 embodiment of the invention, only the center  $N$  samples of the frame of length  $2N$  is  
3 taken, because of the boundary discontinuities, which can be introduced by treating  
4 important low amplitude frequency components as noise and removing them.

5         The nature of these boundary discontinuities can be explained with an example  
6 with reference to an artificial situation, although this discussion is equally applicable to  
7 actual signal situations. If a rectangular window is applied to a fixed non-synchronous  
8 (with respect to the FFT window length) sine wave, a substantial amount of spectral  
9 leakage results. Frequently, this leakage can be seen across all frequency bins, not just  
10 those in bins adjacent or close to the main frequency bin of the sine wave (that closest to  
11 the actual frequency of the sine wave). For the most part, the leakage amplitude is small  
12 compared to that of the main bin, and hence will be removed by the noise reduction  
13 method. Leakage components close to the main bin, however, will generally be larger  
14 and will be masked favorably by the transversal smoothing and will therefore be retained  
15 or only marginally reduced. The resulting frequency plot will appear to be somewhat  
16 similar to that which would be observed had windowing been applied to reduce leakage.  
17 Therefore, when the frequency data is transformed back into the time domain, there is  
18 some amplitude variation at the frame boundaries, the central data being largely  
19 unaffected. For this reason, it is desirable to take only the central data from the processed  
20 frame.

21         Also, it has been observed, that it is possible to use a rectangular window function  
22 on real signals and still get reasonable results from the noise reduction method. This is  
23 generally not the case in other FFT based processing algorithms.

Following the Inverse Windowing 112, the N samples of de-windowed data is passed to the De-emphasis function 113. This De-emphasis function is chosen to undo the frequency emphasis effects of the pre-emphasis function 102. The inverse of the pre-emphasis function 102, described above, a differencing function is used to integrate the data, using the formula:

$$s'(n) = s(n) + s'(n-1);$$

where  $s'(n)$  is the new de-emphasized sample,  $s(n)$  is the current sample to be de-emphasized, and  $s'(n-1)$  is the previous de-emphasized sample. However, due to small errors introduced by using finite precision arithmetic, this integration has a tendency to drift slowly with time, eventually resulting in an overflow situation. To compensate for this drift, a DC blocking function, or high pass filter with a relatively low cut-off frequency, is combined with the integration. The resulting formula is of the form:

$$s'(n) = K * (s(n) + s'(n-1));$$

where K is close to, but less than, 1.0. In the preferred embodiment of this invention a value of 0.984615 is reasonable for K, although other alternative values can be substituted without departing from the concept of this invention.

The N samples of de-emphasized data represents the noise reduced signal and are sent, after de-emphasis 113, to the digital output stream 114.

Figures 2a and 2b are frequency plots, which illustrate the frequency compensation effect of differencing on a speech sample. Figure 2a shows the overall frequency content of a large sample of speech contaminated by road noise. This plot shows about 22 seconds of data sampled at 8 kHz. Figure 2b shows the resulting

1 frequency plot after differencing has been applied. As can quite clearly be seen, the  
2 frequency shape is much flatter after differencing.

3       Figures 3a and 3b are time domain plots showing the time domain effects of pre-  
4 emphasis (differencing) on the waveform. Figure 3a is a time domain plot of a short  
5 sample of speech and noise prior to pre-emphasis. Figure 3b is a time domain plot of the  
6 same short sample of speech and noise after the pre-emphasis function has been applied.  
7 In the preferred embodiment of the invention, differencing is used for pre-emphasis.  
8 Differencing is the simplest pre-emphasis function, although it provides only a rough  
9 approximation of the spectral roll off of the speech signal. In alternative embodiments of  
10 the invention, if a better approximation is required a more complex pre-emphasis  
11 function can be substituted.

12       Figure 4 is a top-level simplified block diagram of buffer handling, showing the  
13 top-level steps of buffer management. In the preferred embodiment of the invention, no  
14 other processing is performed during these steps, other than data movement. First,  
15 samples from the emphasized input stream are stored in an Input Buffer I[n] 401 of size  
16 N, until the Input Buffer 401 is full. This Input Buffer 401 is concatenated with the  
17 Previous Buffer I[n-1] 405, also of size N. The concatenated buffer is copied to the  
18 Working Buffer B[n] 402, of size 2N. The Working Buffer B[n] 402 contains the input  
19 time domain data for the main analysis frame. The buffer concatenation to create a frame  
20 of data in the Working Buffer B[n] 402 provides an effective frame overlap of 50%. That  
21 is, 50% of the data for the current frame is identical to 50% of the data from the previous  
22 frame. Once I[n-1] 405 and I[n] 401 have been copied to B[n] 402, I[n] 401 is moved to  
23 I[n-1] 405 overwriting the previous contents of I[n-1] 405. I[n] 401 is now free to accept

1 further samples from the emphasized input stream. Once the noise reduction process has  
2 been applied to the data in the Working Buffer  $B[n]$  402 to produced the Result Buffer  
3  $R[n]$  403, of size  $2N$ , the central  $N$  samples of  $R[n]$  403 are copied to the Output Buffer  
4  $O[n]$  404, of size  $N$ , for transmission.

5        Figures 5a and 5b are plots of the Hanning and Inverse Hanning Window  
6 function. Figure 5a shows the Hanning Window for an analysis frame of size 128. This  
7 view shows that the Window Function is zero at those endpoints 501, 502 of the window  
8 and near unity at the midpoint 503 of the window. When this Window Function is  
9 applied to the analysis frame, which in this preferred case is also 128 samples in size,  
10 samples 63 and 64 will be essentially unchanged. But moving toward the boundaries  
11 504, 505 of the frame, the samples become increasingly attenuated, to the point where  
12 samples 0 and 127 will be zeroed, irrespective of their original value. This amplitude  
13 modulation of the analysis frame will be present after the signal has been processed in the  
14 frequency domain and is transformed back into the time domain. Since such amplitude  
15 modulation can be undesirable, after processing an inverse function of with Windowing  
16 Function is applied. Because the Windowing Function does not have an inverse for the  
17 end points 501, 502 of the frame, only the central half of the processed (Result) buffer is  
18 used. Figure 5b shows the corresponding inverse function for the Hamming Window of  
19 size 128, for the central half of the function, that is, for samples 32 through 95.

20        Figure 6 is a plot of the typical and preferred weighting function of this invention.  
21 As can be seen for this particular preferred weighting function, bins with smoothed power  
22 levels, above about 47 dB 601, are given a weighting of 1.0, that is, they remain  
23 unchanged. Bins with a smoothed power levels less than about 25 dB 602 are given a

weight of 0.0, that is, they are completely attenuated. Bins with smoothed power levels between about 24 dB and 47 dB 603 are given a weighting between 0.0 and 1.0, with the lower levels having a lower weighting. When normalization is applied, periods of signal that contain only noise may be promoted above the noise cut off levels. If the overall or mean bin power is low, then normalization subtracts less power than when the desired voice components are also present. This tends to give the noise a greater normalized power than desired. To overcome this unwanted side effect of normalization, an absolute weighting may be applied. For example, if the absolute power in a particular bin is less than a particular threshold, a weighting of 0.0 may be applied irrespective of the normalized bin power. A more sophisticated absolute weighting may be applied, such as that for the normalized power. However, it has been observed through experimentation, that a simple absolute cut off threshold gives reasonable results.

The significant improvement that smoothing gives to inter-frame continuity (across the frequency bins) and intra-frame continuity (from frame to frame) is illustrated by example in figures 7a and b and 8a and b. Figures 7a and 7b are process diagrams showing snapshots of a speech sample without the smoothing functions applied. Figure 7a shows snapshots of a first frame at each processing step (input waveform 701, emphasized waveform 702, raw frequency data 703, bin power 704, weighting scalars 705, weighted frequency data 706, emphasized output 707 and output waveform 708), while figure 7b shows snapshots of a consecutive frame at each processing step. In figure 7a, the bin power snapshot 704 shows four regions 704a-d, in the frequency domain, of relatively high power. However, within each of these regions 704a-d there is a great deal of power fluxuation. For this reason the Weighting Scalars, shown in snapshot 705, also



1 fluctuate greatly giving a low degree of intra-frame continuity. Comparing the Bin  
2 Power plot 704 of figure 7a with the Bin Power plot 712 of figure 7b, it is clear that the  
3 overall trend is the same in both plots 704, 712, but these snapshot plots are markedly  
4 different from each other. The Weighting Scalars 705, 713 of figures 7a and 7b  
5 respectively also share this trait, showing a low degree of inter-frame continuity when  
6 smoothing is not applied. Figure 7b also shows snapshot plots of the process steps input  
7 waveform 709, emphasized waveform 710, raw frequency data 711, bin power 712,  
8 weighting scalars 713, weighted frequency data 714, emphasized output 715 and output  
9 waveform 716. These plots, of figure 7b, related to the frame of data, which follows that  
10 of figure 7a.

11 Figures 8a and 8b show snapshots of consecutive frames of a speech sample with  
12 the smoothing functions applied. Again, the snapshot plots of figure 8a are the input  
13 waveform 801, emphasized waveform 802, raw frequency data 803, bin power 804,  
14 weighting scalars 805, weighted frequency data 806, emphasized output 807, and output  
15 waveform 808 of a first frame. While the snapshot plots of figure 8b are the input  
16 waveform 809, emphasized waveform 810, raw frequency data 811, bin power 812,  
17 weighting scalars 813, weighted frequency data 814, emphasized output 815, and output  
18 waveform 816 of a first frame. When smoothing is applied, performing the same  
19 comparison as above regarding figures 7a and 7b, it can be seen that both the Bin Power  
20 804, 812 and the Weighting Scalars 805, 813 show a large degree of intra-frame  
21 continuity, and that the corresponding plots of figures 8a and 8b have only changed  
22 slightly from frame to frame. Smoothing, therefore, enhances both intra-frame continuity  
23 and inter-frame continuity.

Figures 9a-e are spectrograms of a speech sample showing the results of the process of this invention with various processing. These figures further show the benefits of intra and inter-frame continuity. Figure 9a shows a spectrogram of a short sample of speech with car noise. This sample is approximately 2.7 seconds long and was sampled at 8 kHz. The dark areas represent high amplitude frequency components. The lighter the area the lower the amplitude. As can be seen from the lack of white regions, the sample is immersed in a large amount of continuous wide-band noise. Figure 9b shows the result of the processing without smoothing applied. It is clear, by the large regions of white areas, that most of the background noise has been removed. However, the small broken up regions of gray, such as the circled region 903, is quite undesirable. Such narrow frequency components and short duration components are unnatural and can be just as annoying and distracting to the listener as the broadband noise. Figure 9c shows the effect of including temporal smoothing in the processing steps of this invention. Temporal smoothing stretches the energy of the short duration components between frames. When the noise produces an isolated, or short duration component, stretching the component's energy between frames reduces the energy in each frame and, thereby, increases the attenuation applied to the component. Moreover, temporal smoothing eliminates the abrupt cut-off seen in Figure 9b 901 when the frequency bins change from speech to non-speech areas. The circled region 904 has a less abrupt cut-off. Figure 9d shows the effect of including transversal smoothing in the processing steps. In this case, the energy of very narrow, and unnatural, spectral components are stretched between frequency bins, reducing isolated component energy in a particular bin and consequently increasing the attenuation applied to the isolated component. Figure 9e shows the

1 combined effect of including both temporal and transversal smoothing. As can be seen,  
2 the presence of broken up gray regions is greatly reduced. Also, transitions between  
3 speech and non-speech periods 905, with respect to both time and frequency, are less  
4 abrupt and more natural than 902.

5 Figure 10 is a block diagram of the preferred noise reducing apparatus of this  
6 invention, namely a noise-reducing adapter 1001 for a cellular telephone embodiment.  
7 The cellular telephone 1002 is preferably of the type that provides an analogue electrical  
8 signal for the speaker 1003 signal 1012 and accepts an analogue electrical signal 1013 for  
9 the microphone 1004 signal. The noise reducing adapter 1001 provides a connection for  
10 receiving the speaker 1003 signal 1012 from the phone 1002 and, providing that no  
11 further signal amplification is necessary, passes this signal to a connector 1014 that is  
12 compatible with the selected output speaker 1003. The noise-reducing adapter also  
13 provides an input connector 1015 for receiving an analogue signal 1016 from a  
14 microphone 1004. This analogue signal 1016 contains an information component and a  
15 noise component. The analogue signal 1016 is passed to an analogue interface circuit  
16 1011, which amplifies the signal 1016 as necessary, provides the required level of anti-  
17 aliasing filtering, and converts the analogue signal into digital form. The digitized  
18 microphone signal 1017 is received by a digital signal processor 1007, which processes  
19 the signal to reduce the noise component using the noise reducing method previously  
20 described. The program that the DSP 1007 executes is stored in a non-volatile memory  
21 or PROM 1008. The processed digital signal 1018 is passed to interface circuitry 1006,  
22 which converts the processed digital signal 1018 back into an analogue form and  
23 performs any required signal level adjustment prior to transmitting the processed

1 analogue signal to the phone 1002. Additional support circuitry may be required by the  
2 DSP 1007 and the converters 1006, 1011. For example, a clock generating circuit or  
3 crystal 1009 and a power supply and associated conditioning circuitry 1010 are generally  
4 required. The present preferred embodiment of this invention, also has a cigarette lighter  
5 socket 1005 for connected to a car's cigarette lighter socket, in order to provide power for  
6 the adapter 1001. Preferably, the DSP 1007 has on-board volatile random access  
7 memory for storing digital signals and intermediate calculations, as well as signal buffers.

8         The foregoing description is of a preferred embodiment of the invention and has  
9 been presented for the purposes of illustration and description of the best mode of the  
10 invention currently known to the inventors. This description is not intended to be  
11 exhaustive or to limit the invention to the precise form, connections or choice of  
12 components disclosed. Obvious modifications or variations are possible and foreseeable  
13 in light of the above teachings. This embodiment of the invention was chosen and  
14 described to provide the best illustration of the principles of the invention and its practical  
15 application to thereby enable one of ordinary skill in the art to utilize the invention in  
16 various embodiments and with various modifications as are suited to the particular use  
17 contemplated by the inventors. All such modifications and variations are intended to be  
18 within the scope of the invention as determined by the appended claims when they are  
19 interpreted in accordance with the breadth to which they are fairly, legally and equitably  
20 entitled.

## Claims

We claim:

1. A method for reducing unwanted noise in a communication signal, comprising:
  - (A) receiving a digital input stream;
  - (B) pre-emphasizing said received digital input stream producing pre-emphasized data;
  - (C) storing said pre-emphasized data in a buffer;
  - (D) concatenating said buffer containing said pre-emphasized data to produce a frame of data;
  - (E) windowing said frame of data to provide data with a minimum of spectral leakage;
  - (F) transforming said windowed data into the frequency domain;
  - (G) calculating a power estimate for said frequency domain transformed data;
  - (H) temporally smoothing said power estimate to produce time smoothed data;
  - (I) transversally smoothing said time smoothed data to produce smoothed power data;
  - (J) weighting frequency values based on said smoothed power data to provide weighted FFT data;
  - (K) inverse transforming said weighted FFT data to provide a time domain waveform;
  - (L) inverse windowing said time domain waveform to provide a de-windowed time domain sample;

1 (M) de-emphasizing said de-windowed time domain sample to remove  
2 frequency emphasis effects from said time domain sample; and  
3 (N) generating a digital output stream of said de-emphasized data.

4 2. A method for reducing unwanted noise in a communication signal, as recited in  
5 claim 1, wherein said received digital input stream originates from a cellular telephone  
6 having a digital voice output.

7 3. A method for reducing unwanted noise in a communication signal, as recited in  
8 claim 1, wherein said pre-emphasizing flattens the spectral energy of said received digital  
9 input stream.

10 4. A method for reducing unwanted noise in a communication signal, as recited in  
11 claim 1, wherein said concatenating said buffer, further comprises combining a previous  
12 input buffer with said buffer to provide a frame overlap of approximately 50%.

13 5. A method for reducing unwanted noise in a communication signal, as recited in  
14 claim 1, wherein said windowing employs a Hanning Window function.

15 6. A method for reducing unwanted noise in a communication signal, as recited in  
16 claim 1, wherein said windowing employs a Rectangular Window function.

17 7. A method for reducing unwanted noise in a communication signal, as recited in  
18 claim 1, wherein said transforming further comprises using a Fast Fourier Transform to  
19 create one or more resulting frequency domain data frequency bins.

20 8. A method for reducing unwanted noise in a communication signal, as recited in  
21 claim 1, wherein said calculation of power estimate further comprises summing the  
22 squares of the real components of each frequency bin to the squares of the imaginary  
23 components of each frequency bin.

1 9. A method for reducing unwanted noise in a communication signal, as recited in  
2 claim 1, wherein said temporally smoothing further comprises averaging said power  
3 estimate.

4 10. A method for reducing unwanted noise in a communication signal, as recited in  
5 claim 1, wherein said temporally smoothing further comprises low pass filtering said  
6 power estimate.

7 11. A method for reducing unwanted noise in a communication signal, as recited in  
8 claim 1, wherein said transversely smoothing further comprises averaging said time  
9 smoothed data.

10 12. A method for reducing unwanted noise in a communication signal, as recited in  
11 claim 1, wherein said transversely smoothing further comprises low pass filtering said  
12 time smoothed data.

13 13. A method for reducing unwanted noise in a communication signal, as recited in  
14 claim 1, wherein said weighting frequency values further comprises:

- 15 (1) generating an array of weighting scalars; and  
16 (2) multiplying said array of weighting scalars by said frequency  
17 domain transformed data.

18 14. A method for reducing unwanted noise in a communication signal, as recited in  
19 claim 1, wherein said inverse transforming uses an Inverse Fast Fourier Transform.

20 15. A system for reducing unwanted noise in a communication signal, comprising:

- 21 (A) a telephone;  
22 (B) a noise reducing telephone adapter in electronic communication with said  
23 telephone;

1 (C) a speaker in electronic communication with said noise reducing telephone  
2 adapter; and

3 (D) a microphone in electronic communication with said noise reducing  
4 telephone adapter.

5 16. A system for reducing unwanted noise in a communication signal, as recited in  
6 claim 15, wherein said noise reducing telephone adapter, further comprises:

7 (1) a processor;

8 (2) an analogue to digital converter electrically connected to said  
9 processor;

10 (3) a digital to analogue converter electrically connected to said  
11 processor; and

12 (4) a memory unit electrically connected to said processor.  
13



## **Abstract**

A method and system for reducing the undesirable noise in a communication signal is provided. Designed specifically to address the problem of telephone communications where the desired speech signal is contaminated by background noise, this invention employs digital signal processing of the communication signal to selectively emphasize, buffer, amplify, and smooth the components of the signal, thereby enhancing the signal quality (signal to noise ratio) of the presented communication signal.

FIG. 1 is a block diagram of a digital signal processing system.

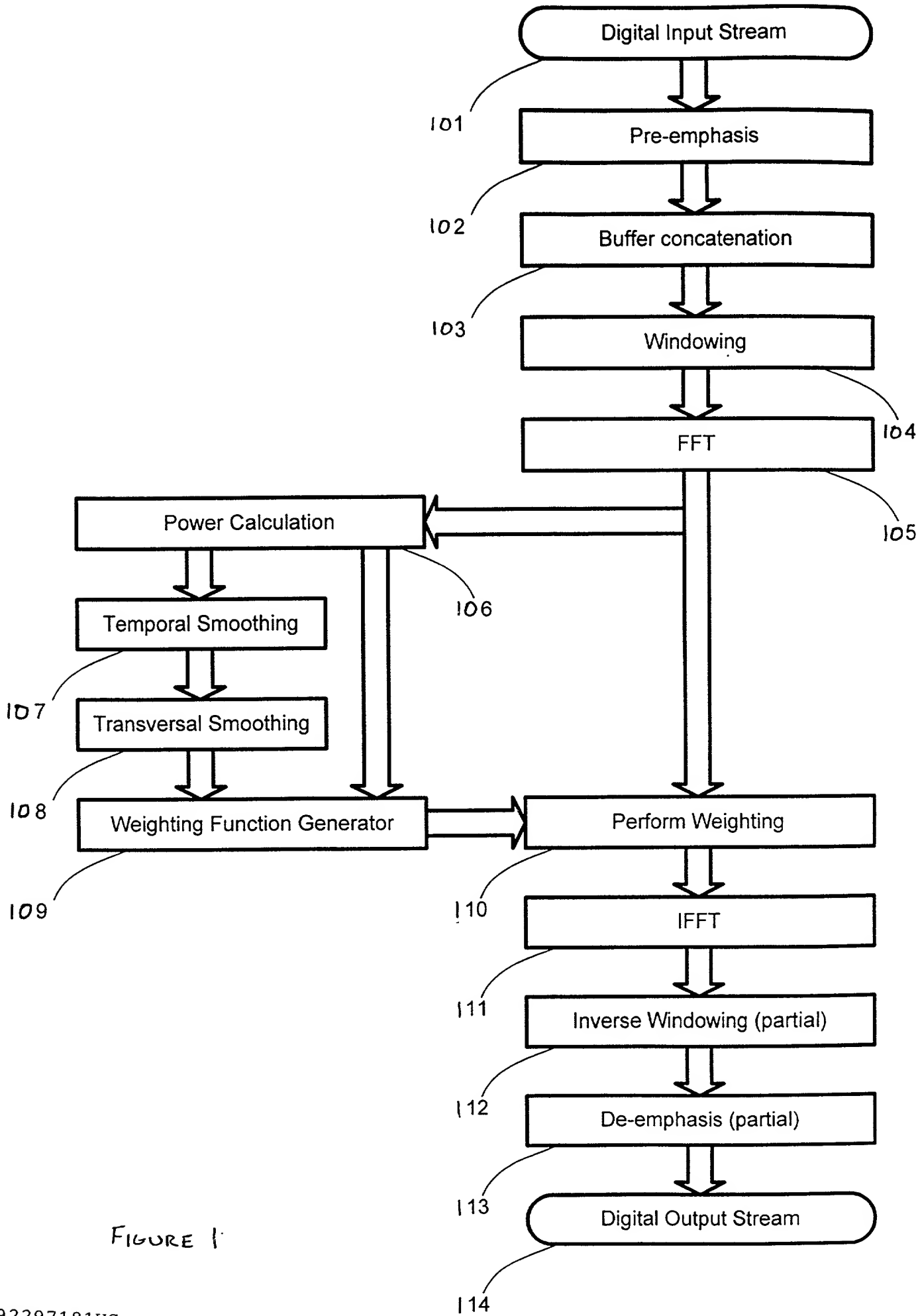


FIGURE 1

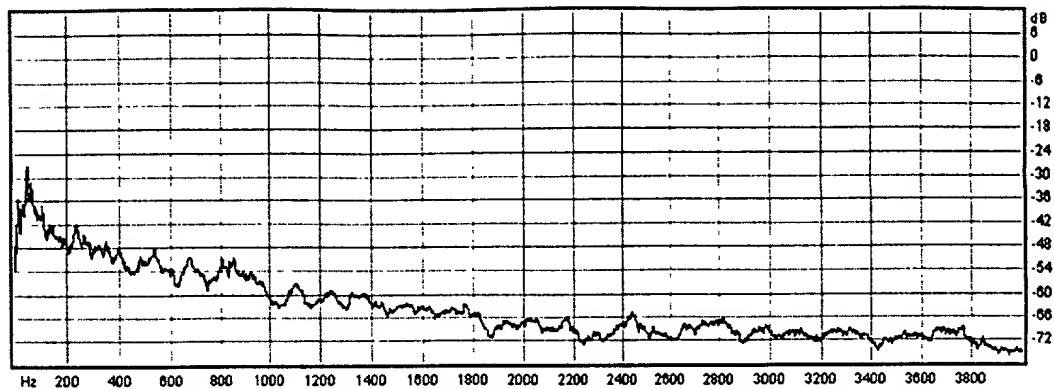


Fig 2a: Frequency plot of large sample of speech and noise.

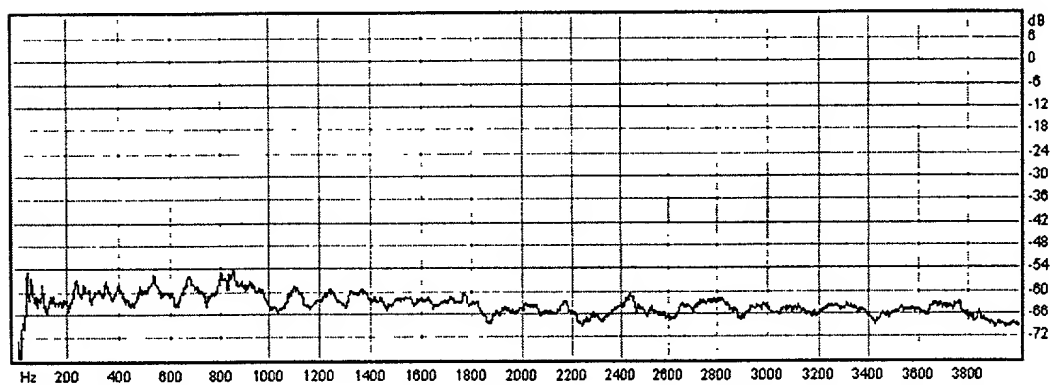


Fig 2b: Frequency plot of large sample of speech and noise after emphasis function has been applied.

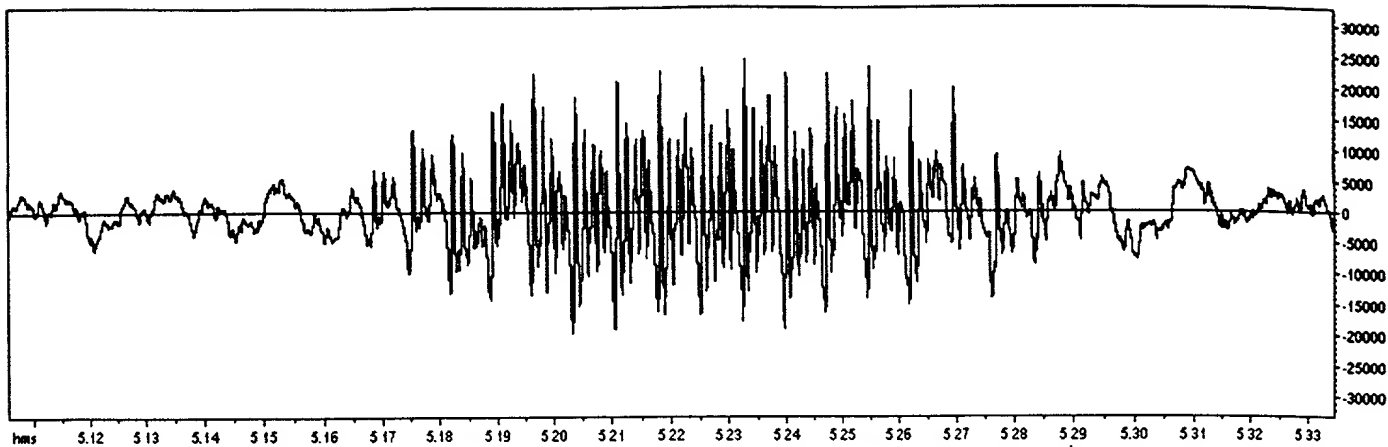


Fig 3a: Time domain plot of short sample of speech and noise.

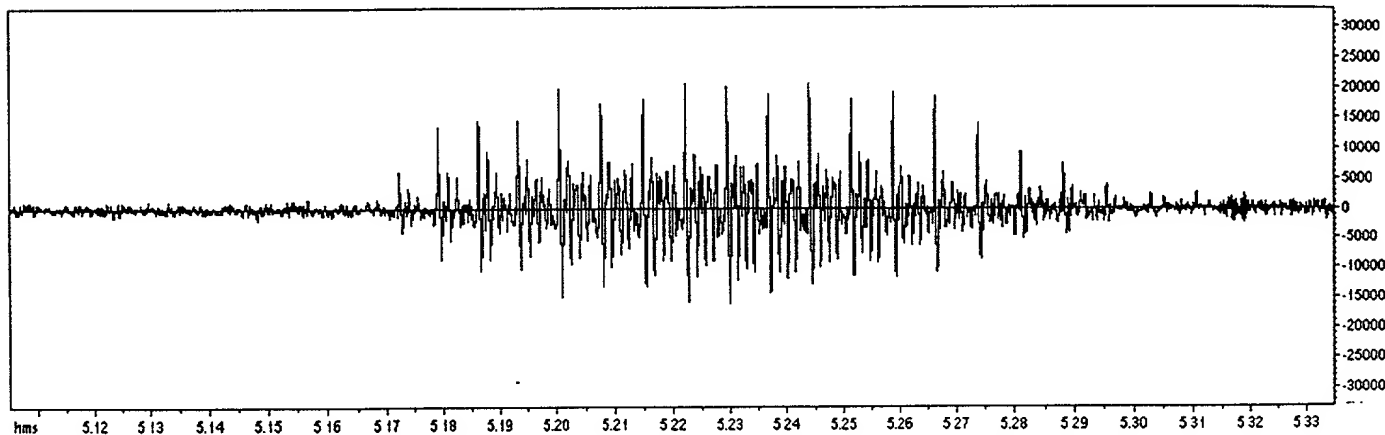
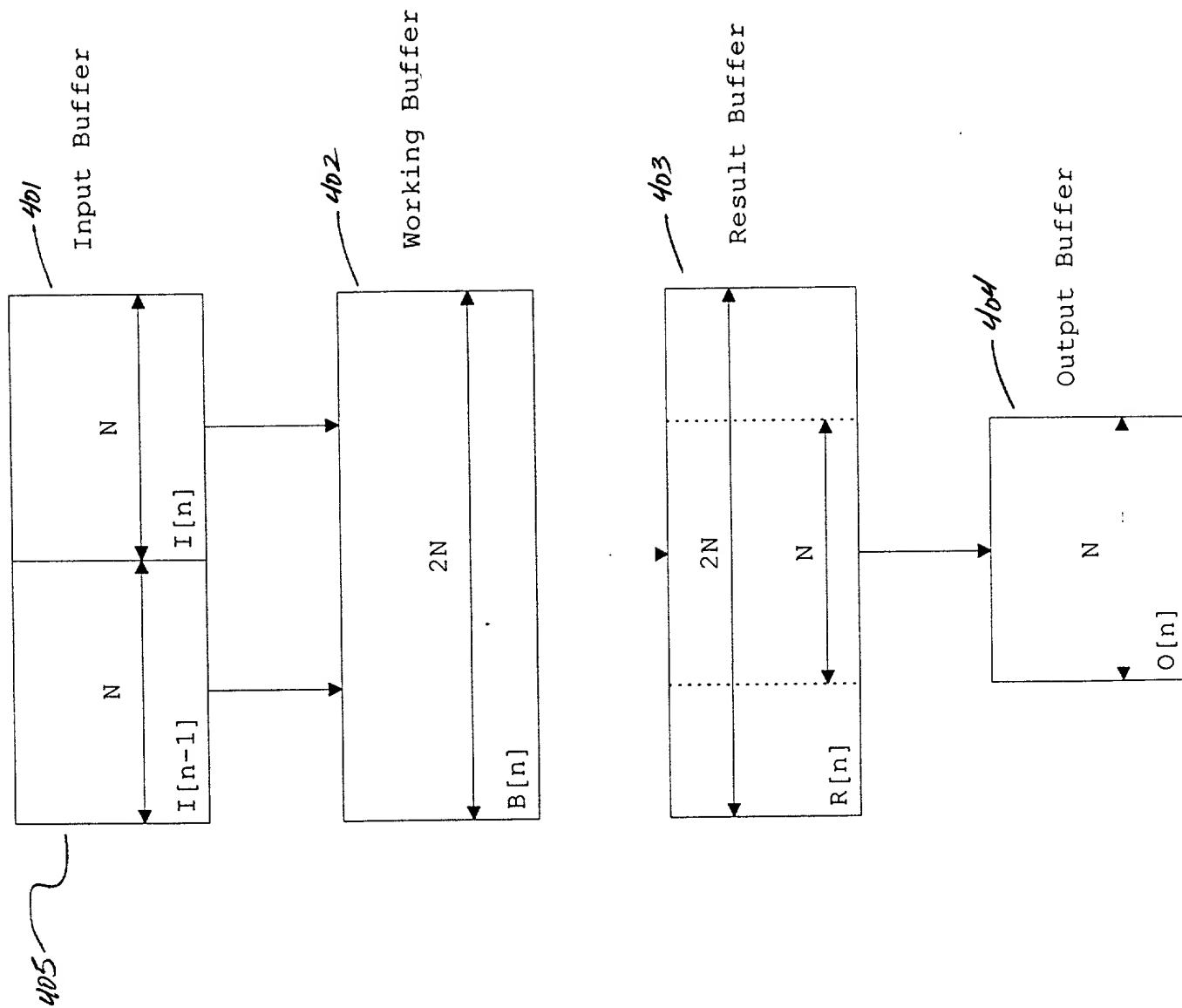


Fig 3b: Time domain plot of short sample of speech and noise after emphasis function has been applied.

FIG. 4 is a block diagram of a system 400 for processing data. The system 400 includes an input buffer 401, a working buffer 402, a result buffer 403, and an output buffer 404. The input buffer 401 receives data from a source 405 and stores it in two segments, each of size N. The first segment contains data I[n-1] and the second segment contains data I[n]. The input buffer 401 outputs data to the working buffer 402. The working buffer 402 has a total size of 2N and stores data B[n]. The working buffer 402 outputs data to the result buffer 403. The result buffer 403 has a total size of 2N and stores data R[n]. The result buffer 403 outputs data to the output buffer 404. The output buffer 404 has a total size of N and stores data O[n].



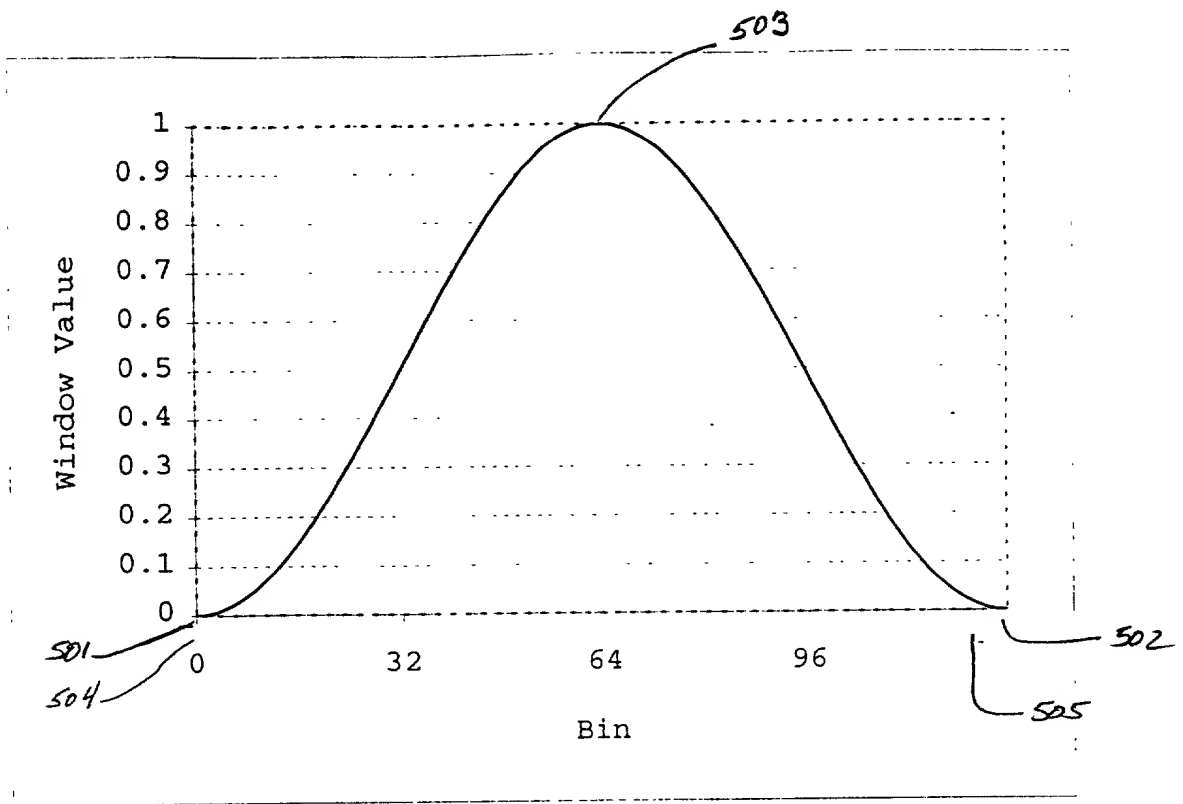


Fig 5a: Hanning Window Function

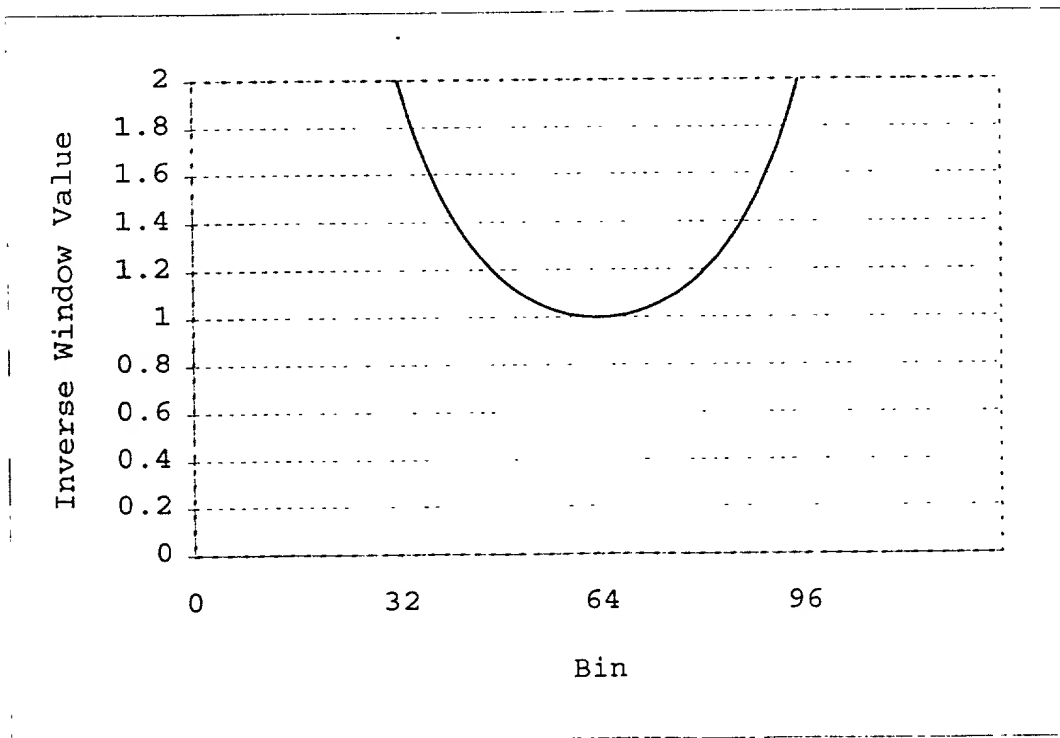


Fig 5b: Inverse Hanning Window Function

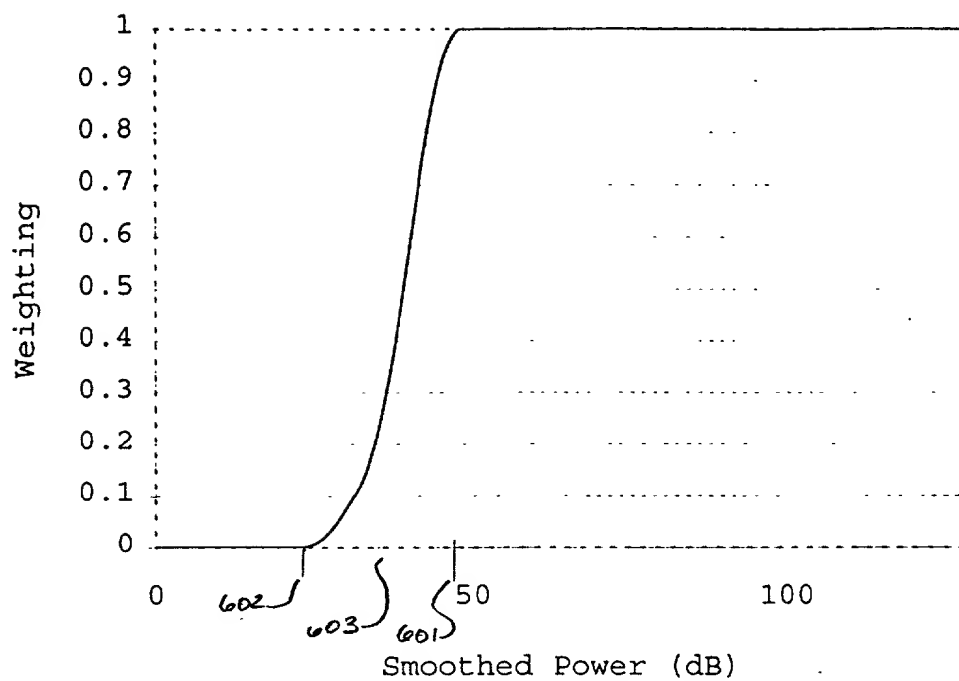
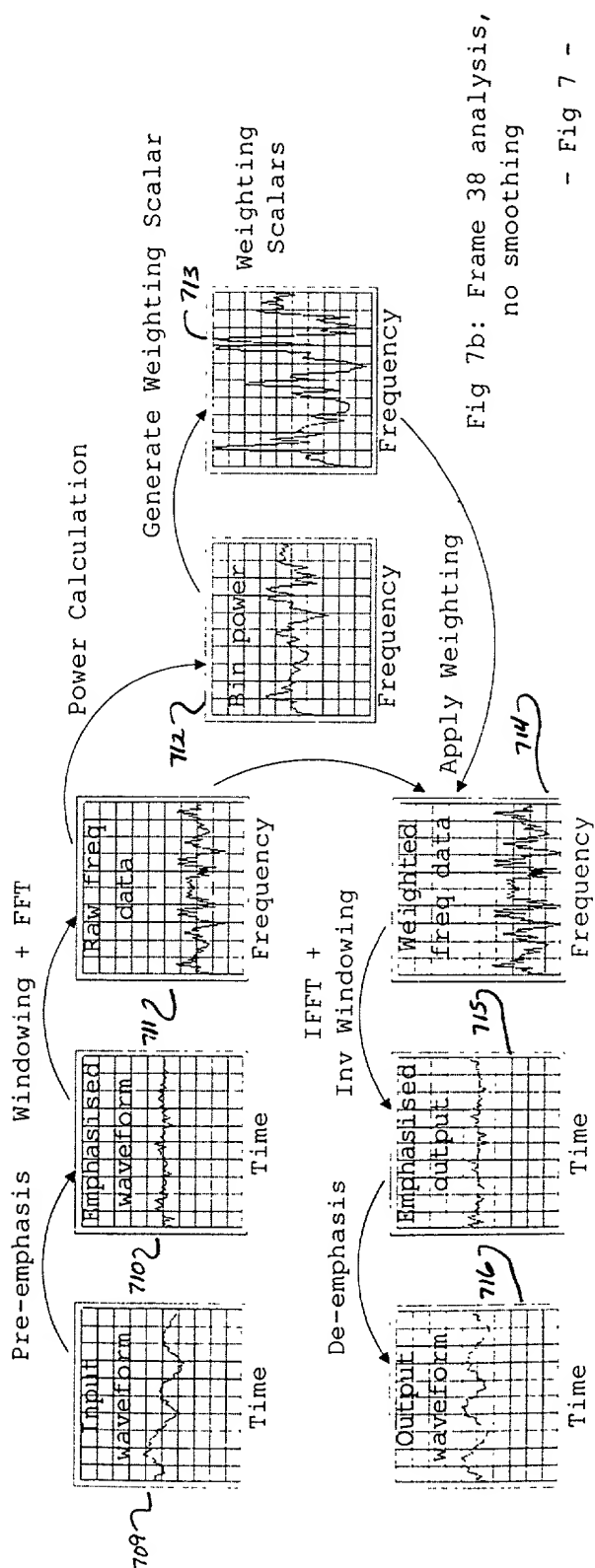
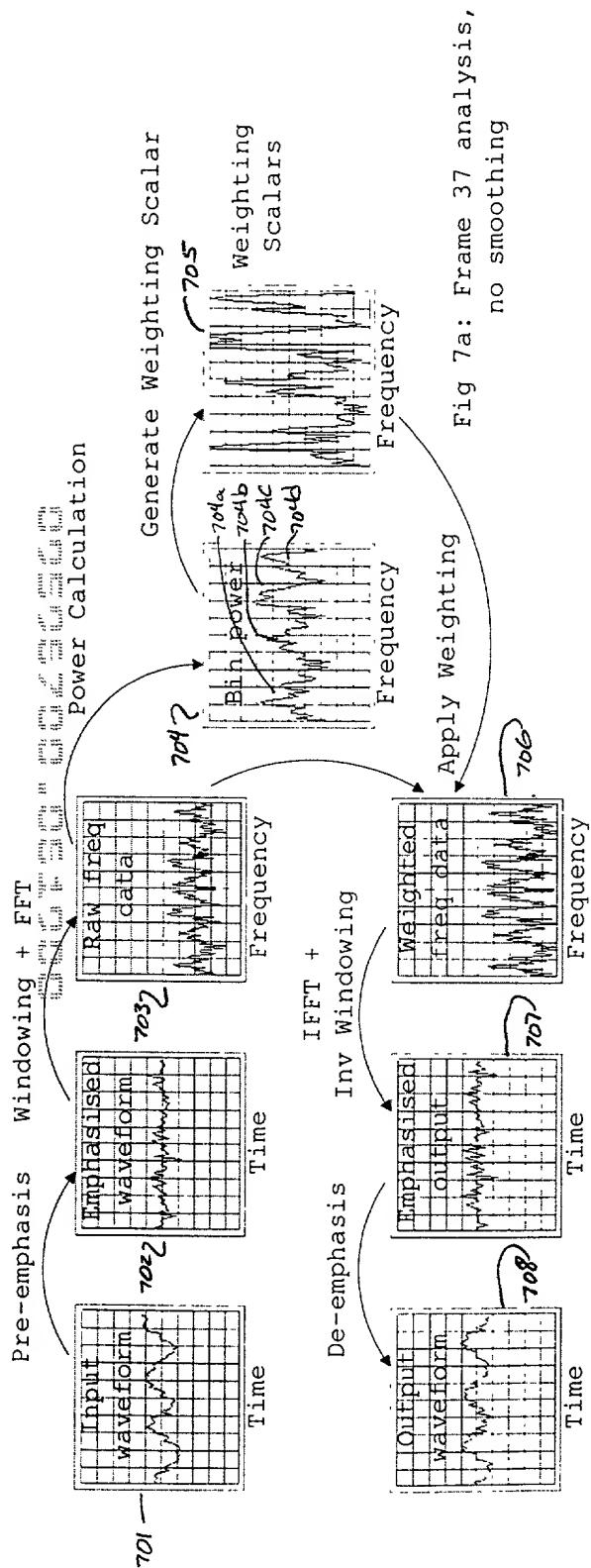


Fig 6: A Typical Weighting Function





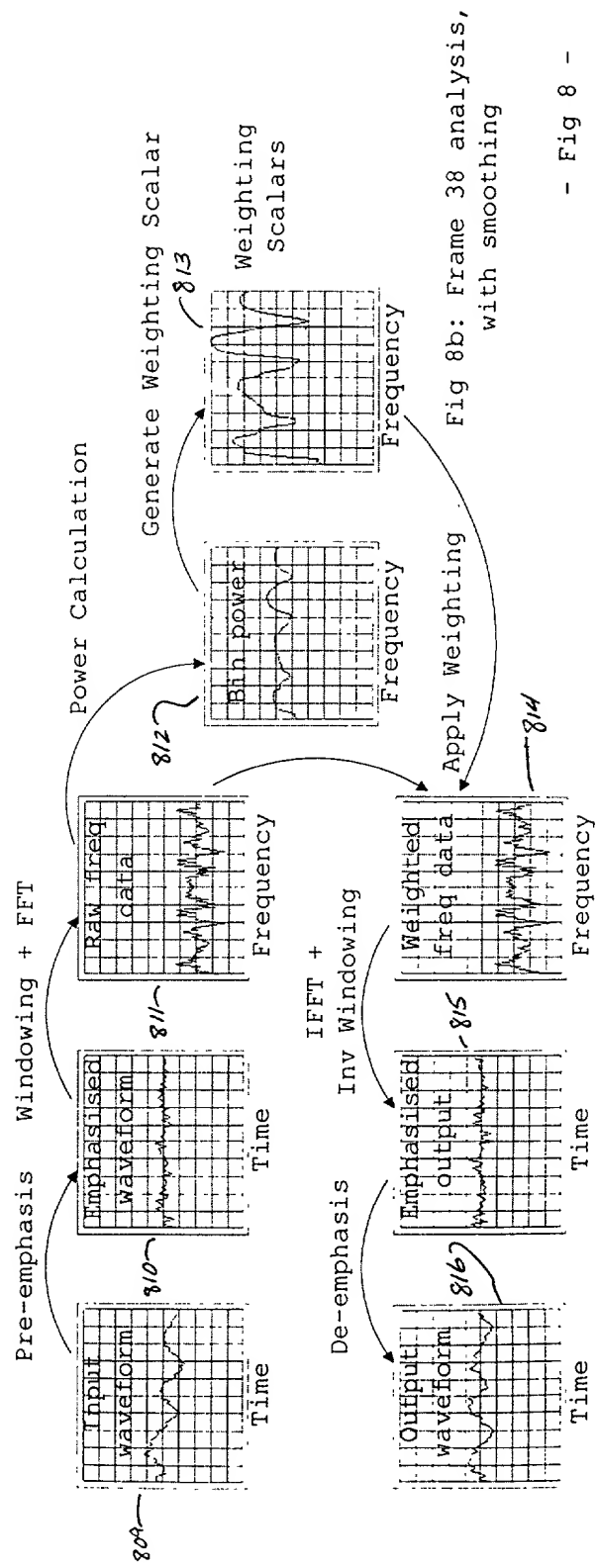
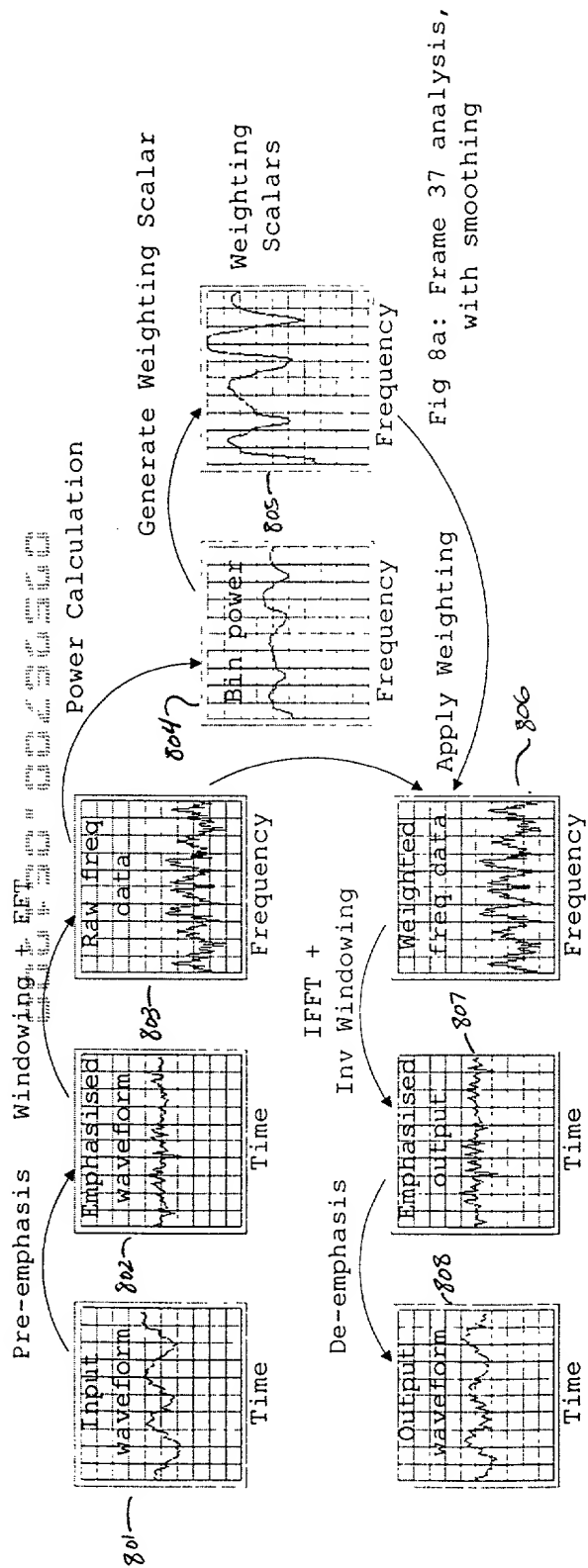


Fig 9a: Input signal

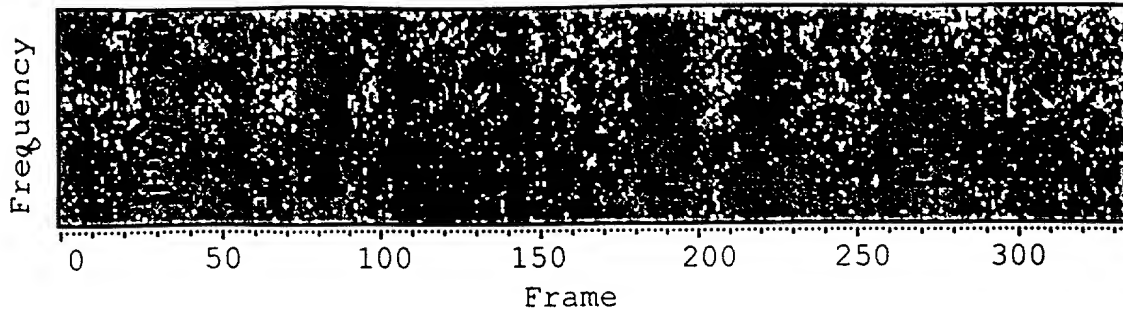


Fig 9b: Processed signal with no smoothing

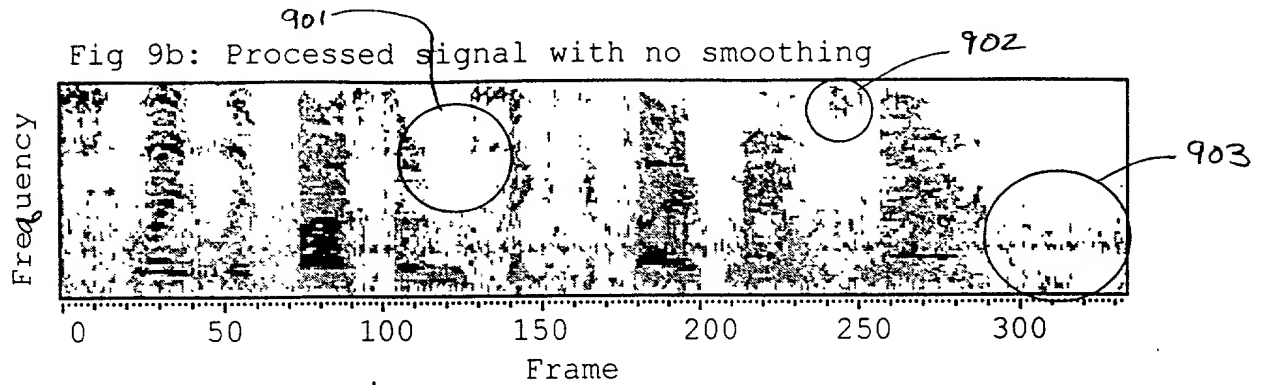


Fig 9c: Processed signal with temporal smoothing only

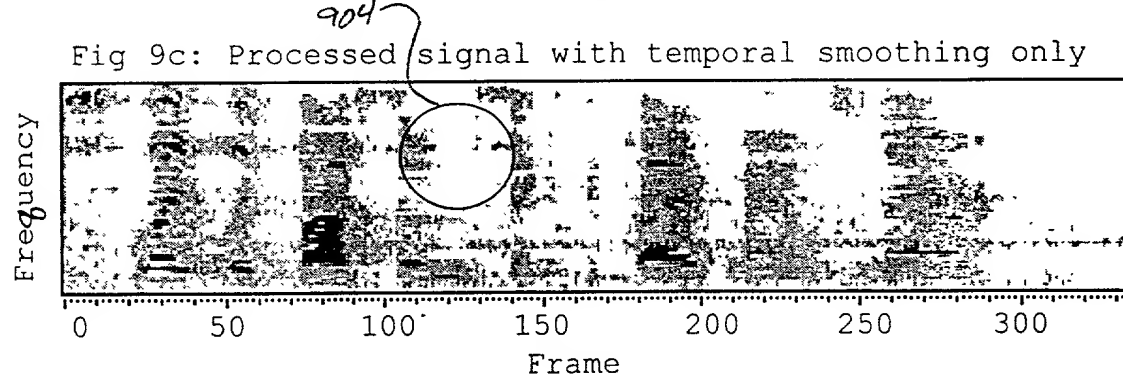


Fig 9d: Processed signal with transversal smoothing only

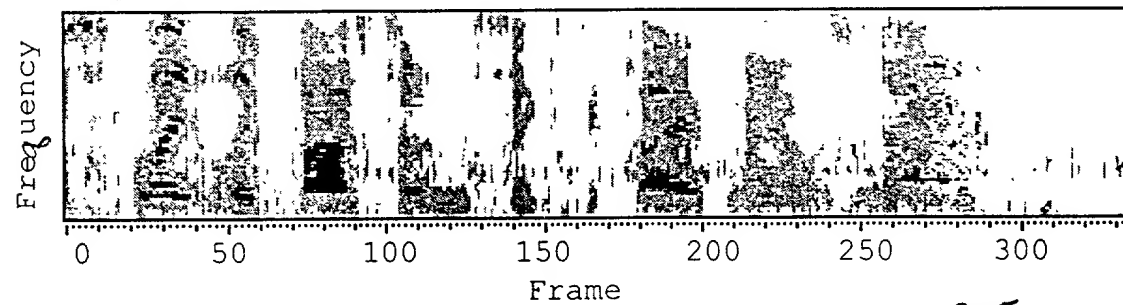
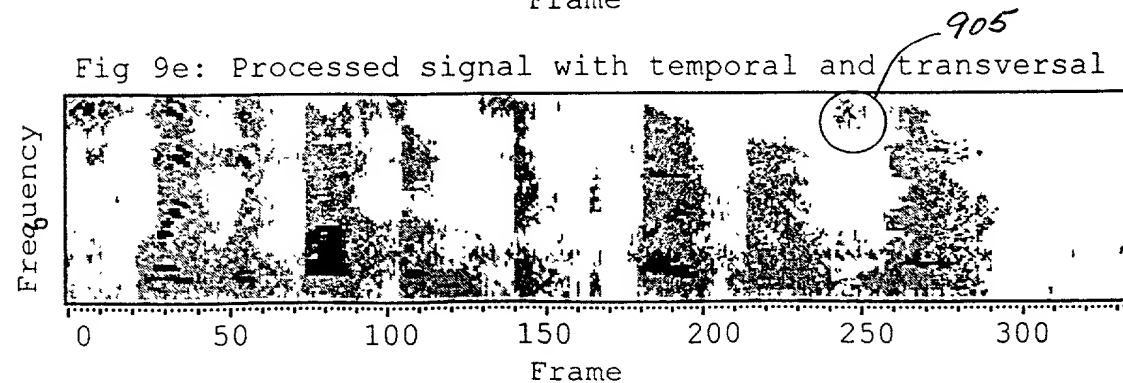
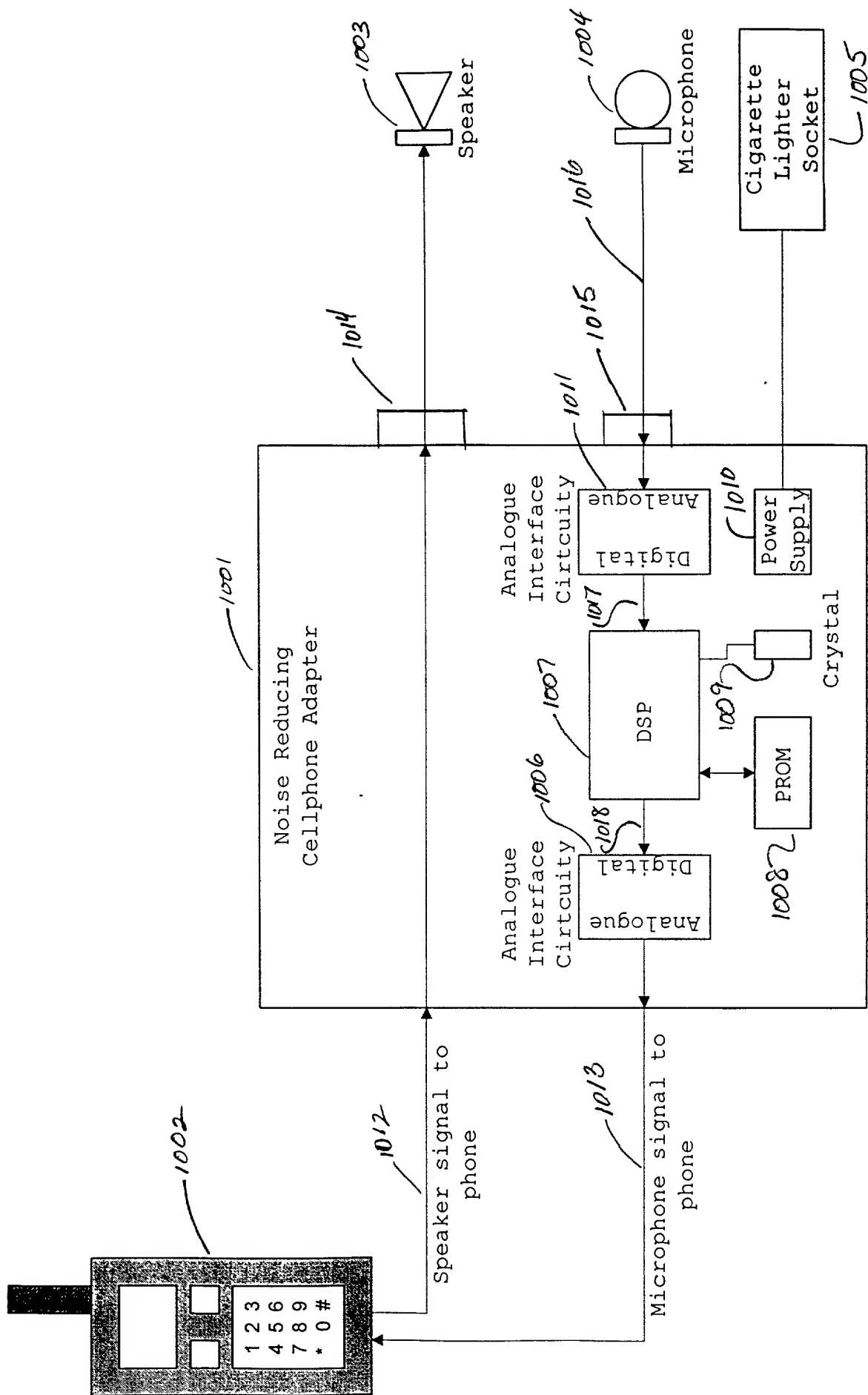


Fig 9e: Processed signal with temporal and transversal smoothing





- Fig 10 -

06/19/00



Jc848 U.S. PTO

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

Jc759 U.S. PTO

09/596700

06/19/00

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James M. Turnbull

ASSIGNEE: Jabra Corporation

SERIAL NUMBER: n/a

DATE FILED: n/a

TITLE: NOISE REDUCTION METHOD AND APPARATUS

ATTORNEY DOCKET: 4246 P

Assistant Commissioner for Patents  
Washington, DC 20231

**DECLARATION FOR PATENT APPLICATION**

Honorable Assistant Commissioner:

As the below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe that I am an original, first and joint inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled **NOISE REDUCTION METHOD AND APPARATUS** the specification of which

☒ is attached hereto

☐ was filed on \_\_\_\_\_

as Application Serial No. \_\_\_\_\_

and was amended on \_\_\_\_\_ (if applicable).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, § 1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, § 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that

of the application on which priority is claimed:

Prior Foreign Application(s)

\_\_\_\_\_  
(Number)

\_\_\_\_\_  
(Country)

\_\_\_\_\_  
(Date)

Priority Claimed

☐ Yes ☐ No

\_\_\_\_\_  
(Number)

\_\_\_\_\_  
(Country)

\_\_\_\_\_  
(Date)

☐ Yes ☐ No

\_\_\_\_\_  
(Number)

\_\_\_\_\_  
(Country)

\_\_\_\_\_  
(Date)

☐ Yes ☐ No

I hereby claim the benefit under 35 U.S.C. § 119(e) of any United States provisional application(s) listed below.

Prior Provisional Application(s)

\_\_\_\_\_  
(Application No.)

\_\_\_\_\_  
(Filing Date)

\_\_\_\_\_  
(Application No.)

\_\_\_\_\_  
(Filing Date)

\_\_\_\_\_  
(Application No.)

\_\_\_\_\_  
(Filing Date)

I hereby claim the benefit under Title 35, United States Code, § 120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, § 112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, § 1.56(a) which occurred between the filing date of the prior application and the national or PCT international filing date of this application:

Prior United States Application(s):

\_\_\_\_\_  
(Application No.)

\_\_\_\_\_  
(Filing Date)

\_\_\_\_\_  
(Status - patented, pending, abandoned)

\_\_\_\_\_  
(Application No.)

\_\_\_\_\_  
(Filing Date)

\_\_\_\_\_  
(Status - patented, pending, abandoned)

\_\_\_\_\_  
(Application No.)

\_\_\_\_\_  
(Filing Date)

\_\_\_\_\_  
(Status - patented, pending, abandoned)

### Power of Attorney

I hereby appoint Lloyd W. Sadler (Reg. No. 40,154) and Daniel P. McCarthy (Reg. No. 36,600) as my representatives and attorneys and/or agents to prosecute this application and to

transact all business in the Patent and Trademark Office connected therewith. All communications should be directed to Mr. Sadler at the following address or telephone number:

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(801) 323-9399 - telephone  
(801) 323-9396 - facsimile

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

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Inventor's Signature:

Marcia R. Brumitt

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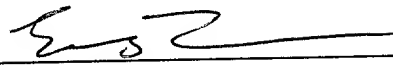
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